

FEB Calibration & DSP Issues J.Oliver, N.Felt

What can we learn from placing FEB in "DSO" mode?

- "DSO" mode → Take continuous samples of baseline at 500 ns/sample.
- 500 us total or, 1k data points @ 2 bytes per point = 2k bytes per channel
- 64k bytes per FEB

Answer

- Preamplifier noise voltage density, e_n
- \bullet Equivalent APD leakage current, \boldsymbol{I}_{L}
- Continuity check → No noise means no continuity
- Optimization of FIR filter coefficients
- Minimization of noise "hit rates"



To study – Create realistic noise traces using known sources, then apply filtering techniques

- Parallel noise
 - \triangleright Source : leakage current shot noise \rightarrow I_L
 - ➤ Integrated by preamp (integrator) then filtered by DCS

$$N_{enc}(I) := \sqrt{\frac{I \cdot \Delta T}{q}}$$

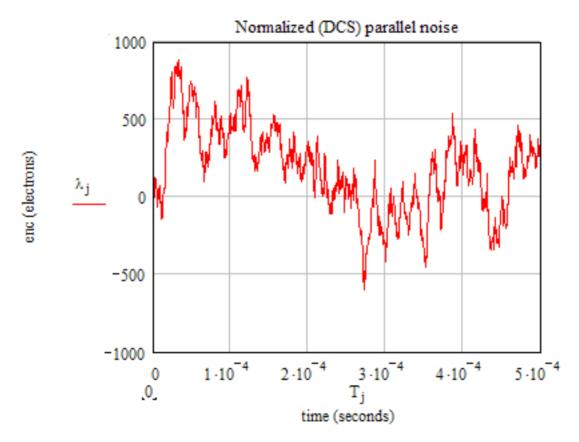
- ➤ Start with Gaussian white noise sample (oversampled at 16 MHz) 8k points
- ➤ "Integrate" with preamp → replace circuit eqn with finite difference equation → recursion formula

$$\Gamma 0_{i} := \begin{bmatrix} 0 & \text{if } i = 0 \\ & & \\ \hline \frac{T_{f}}{T_{f} + T_{s}} \cdot \Gamma 0_{i-1} + \frac{T_{f}T_{s}}{T_{f} + T_{s}} \cdot U1_{i} & \text{otherwise} \end{bmatrix}$$

- ➤ Downsample to 2 MHz (500 ns) samples
- ➤ Apply DCS and normalize to expected value (above) for 1nA leakage



Noise trace corresponding to 1 nA (equivalent) leakage current and $\Delta T = 2us$ sampling (integration) time (~ MASDA)



Note: For the current purposes, we assume digitization @ 50e/count

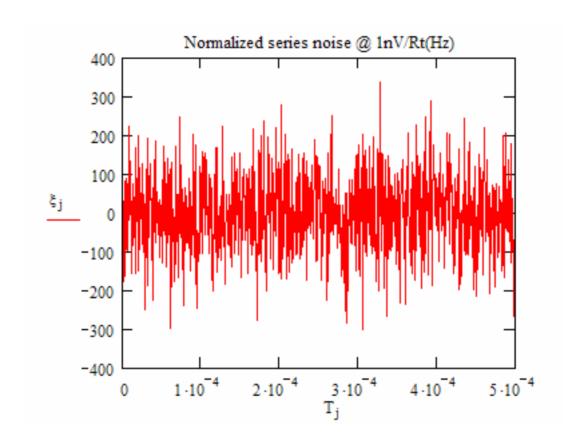


Series noise trace

- Source → Thermal noise from preamp input FET
- Characterized by noise spectral density e_n (in range 1 2 nV/rt(Hz))
- Filtered by shaper risetime constant, Tr (in range 100ns 500ns)
- Again take white Gaussian noise sample
 - > Apply shaper transfer function as finite difference equation
 - \triangleright Normalize to DCS @ $\Delta T = 2 \mu s$
 - \triangleright For en=1 nV/rt(Hz) \rightarrow

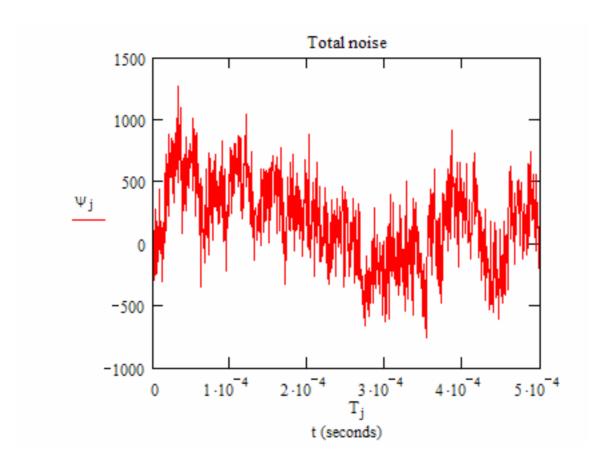
$$ENC := \frac{1nVperRtHz}{q} \cdot \frac{C_d}{\sqrt{2T_r}}$$







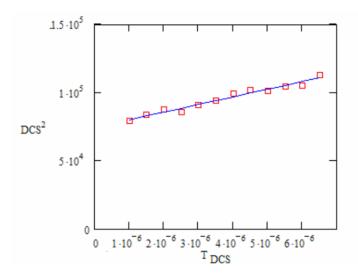
Total noise @ 1 nA leakage, 1 nV/rt(Hz) FET noise, & 100 ns shaping time



- For different values of leakage and en, just scale the terms appropriately
- For different shaping time, redo the recursion formula for series noise



- To recover leakage current
 - \triangleright Compute enc as function of ΔT from 1us 10 us
 - > enc^2 should be proportional to leakage current & ΔT
 - \triangleright plot enc² vs Δ T and find slope



➤ Recovers leakage current to ~ 10%



To recover FET thermal noise density. This is a little harder and we can use one of two methods.

Method #1

- Take multiple baseline samples, each with a different shaping time.
- \triangleright Plot enc² vs (1/Tr) and compute the slope.
- Extract en from slope Note: We must assume detector capacitance is known
- ➤ Shaping time is ASIC programmable : We don't know exactly its value.

Method #2

- Take single baseline sample at 16 MHz with minimum risetime constant.
- ➤ Shape the risetime constant with digital filter. We can do this if the sampling time is small
- \triangleright Plot enc2 vs (1/Tr) and compute the slope.
- ➤ In this case, the shaping time is known better since it's inserted into the DSP filter



FIR data filters

- In general, it's a vector of coefficients α_k
- It operates on the ASIC output stream as follows

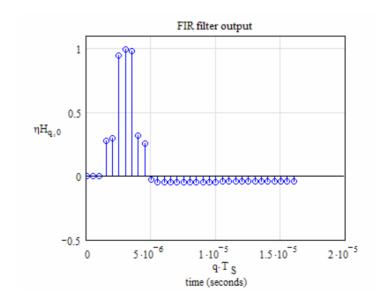
$$\psi_n = \sum_k \alpha_k \cdot \psi_{n-k}$$

- In this case, k corresponds to 500 ns samples
- For DCS

$$\alpha = \begin{bmatrix} -1 \\ 0 \\ 0 \\ 1 \\ 0 \\ 0 \\ 0 \end{bmatrix}$$

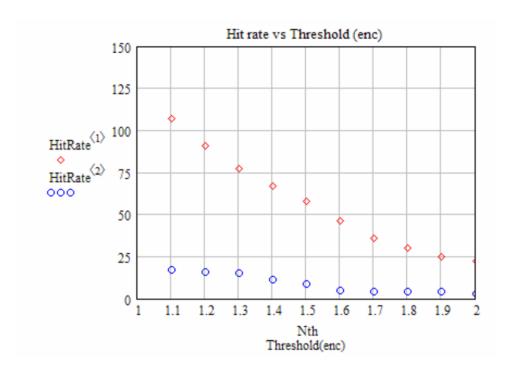


- In general, we can use as many coefficients as we want
- Coefficients are optimized by operating on real noise traces extracted from FEB
- How much we win by this depends on the noise
 - For high e_n and low leakage, we win a lot
 - For low e_n and high leakage, we win less
- Secondary benefit of multiple samples can be seen by looking at signal output. For quad sampling, real signal output looks like;



- Note that filter output is high for 3 consecutive bins
- If we trigger only when signal is over threshold for two consecutive bins, this drastically reduces the spurious noise hit rate.
- Filter output is also quite independent of phase.





HitRate <1> = Transitions with 1 or more bins over threshold HitRate <2> = Transitions with 2 or more consecutive bins over threshold



Data transport issues

- Assume DSO mode takes place at 1 MB/s (should be easy with 16 MHz clock)
- 500 us of baseline @ 2 Msps → 1k data points
- Assume 2 bytes per point \rightarrow 2 kB per channel
- 2 ms transport time
- x32 ch = 64 ms to acquire all channels assuming no bandwidth limitation in concentrator
- 2 kB (16kb) buffering required on FEB.
- Spartan 3E family 72-376 kb Block RAM



Summary

DSO mode features

- Extract noise parameters of preamp and APD
 - ➤ Empirical construction of FIR filter coefficients or
 - ➤ Measure noise autocorrelation function
 - > Compute optimum FIR filter coefficients
 - ➤ Routines can be supplied to DAQ (N.Felt/MatLab) to use as part of calibration routines
- Detect dead channels (ASIC and APD)
- Detect connectivity problems ASIC to APD
- Short and open detection
- Once noise measurements are made, we expect them to change little over time
- This is more like a "once a week" operation than a "once per minute" one.